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Speech Processing

Tony Jones G7ETW covers the techniques of speech tailoring and processing.

Look at almost any modern HF radio, and you will find it has some kind of audio tailoring to give SSB transmissions 'more punch'. These 'speech processor' features are easy to use but I think the subject itself is confusing. To explain, I need to start at square one.

Power in a Transmission

A radio transmitter at its simplest generates RF energy of fixed amplitude on a single 'carrier' frequency. Information is conveyed by switching the transmitter on and off in a recognised mark-space pattern called CW or 'Morse' code, named after its inventor **Samuel Finley Breese Morse**. (Finley Breese? 'Fine Business' – no, surely it's not possible...)

After CW came Amplitude Modulation (AM), in which the carrier's amplitude varies with that of the audio being transmitted. The carrier has no information in it and the audio is duplicated in two 'sidebands'. In fact, most of the transmitted power is wasted. For broadcast use this doesn't matter because radio stations can put out hundreds of kilowatts but for communications a more efficient mode was needed.

Out of AM, in 1947, came 'Single Sideband' (SSB) in which the carrier and one sideband are not transmitted. The remaining sideband is where all the power goes and carries the information.

Amateurs talk of transmitters in terms of their output, usually the maximum. *"I'm running 100W"* we say. When using CW, this is true – the meter shows a constant 100W on key-downs. But on SSB the output follows the operator's voice and rarely gets anywhere the maximum possible. This, finally, brings me to 'speech processors', which are a way to increase the average level of the audio and hence the drive to the PA, resulting in more of the available power being used.

See **Fig. 1** for CW, AM and SSB representations showing power distributions and efficiency (in the sense of power utilisation for the conveying of actual information).

There are two ways to increase average speech level, and manufacturers use both. These methods are compression and clipping, and each can be done at audio or RF frequencies. I'll start with audio – it's easier to visualise.

Audio Tailoring

In a compressor, an audio signal is amplified but not evenly. When the incoming level is low the gain is high, and at higher signal levels little or no gain is applied. This obviously increases the average amplitude by boosting quiet signals. But there is problem with this. When the level is very low such as during periods of near-silence, the gain is huge and the transmitted audio can have very noticeable background noise from sources such as power-supply fans, ticking clocks and in my case once, a cockatiel singing in the next room!

To solve this problem, compressors usually incorporate a 'downward noise gate', which sets the minimum level at which there should be any output at all. The output of a compressor when properly set up is just the information to be transmitted, optimised for both low and high levels.

A clipper works in a different way. The audio is amplified but evenly this time. The resultant waveform is then truncated or 'clipped', to remove the top of the waveform, making the output more like a square wave. This increases the average amplitude by holding the loud signals back. This sounds crude in comparison to a compressor but has one advantage – a clipper has no 'memory'. That is to say, the gain of the amplifier does not have to ramp up and down as the incoming audio varies.

Now look at **Fig. 2**. This is an audio clipper from a 1983 G-QRP Club *Circuit Handbook*. This, the author said, allowed him to *"sit a couple of feet from the microphone and smoke his pipe"* while operating. The circuit's operation depends on the two diodes just before the output, which shunt signal above their switch-on voltages to ground. This results in a waveform of about 300mV peak-to-peak; OA91s are germanium diodes. Modern

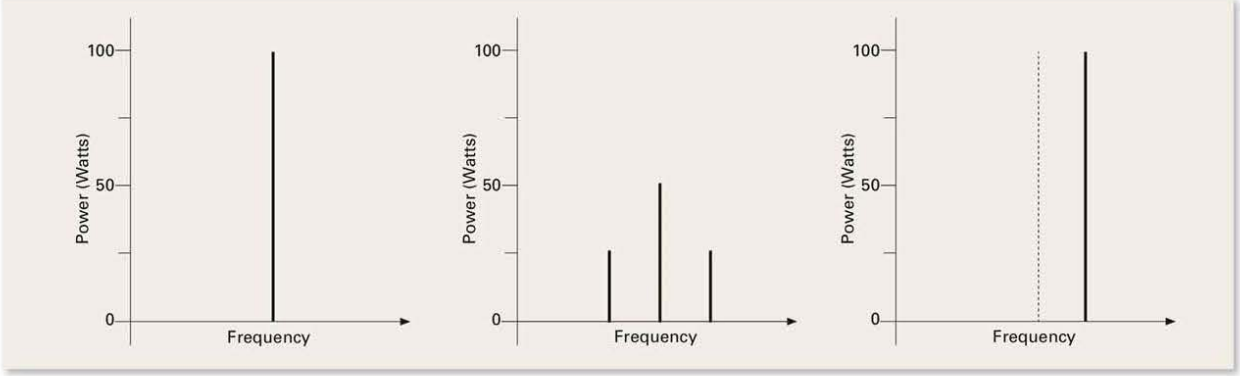


Fig. 1: Power distribution and efficiency for CW, AM and SSB.

silicon diodes could be used but the gain would need to be adjusted.

Fig. 3 shows a more modern audio circuit, a compressor this time. The SM2165 offers 18dB of compression. At the desired signal level and up to 1dB over it, no gain or attenuation is applied but above that, up to 17dB of attenuation is automatically applied. This is best explained with an example. Suppose a microphone preamplifier has a desired maximum output of 200mV but with loud voices it can peak at 500mV. A 'compression ratio' of 5dB means that all but 1dB of the top 5dB is thrown away. So, given an input of 500mV (–3dBV), the output would be –3dbV less a further 4dbV, which is –7dbV or 200mV.

The SSM2165 is obsolete (although they are still available but have been superseded twice, by the 2166 and, currently, the 2167) but, in any case, I have found a modern chip that is also a compressor. See **Fig. 4** for an example application of a MAX9814 chip, taken from a manufacturer's datasheet. This chip is only available in an smd (surface mount) package but there is no need to make your own board because it is commonly used on electret microphone boards such as the 'Adafruit AGC Electret Microphone Amplifier MAX9814' for use with Arduinos and the like. These run off 5V, which is very handy. Configuration is done by use of external resistors and capacitors.

Clipping at audio frequencies generates a new waveform that has distortion comprising odd-number harmonics. Within a frequency range of 2.7kHz, this is a problem because the third harmonics in a human voice can easily fall into this range and a lowpass filter cannot attenuate them.

There is a way to solve this problem – the use of 'split bands'. The incoming audio is split into narrow frequency bands and each is amplified and processed in the usual way. Because each sub-band is narrow, the odd-number harmonics now fall outside and can be removed with a suitable filter. The processed sub-bands must be re-added to make a processed waveform, of course. This, despite being obviously more complex, is the method used by KG4JJH in his VocalMaster speech processor, which he developed for use with Yaesu FT-817s.

Radio Frequency Audio Tailoring

Moving away from audio-frequency designs, we get to what was considered

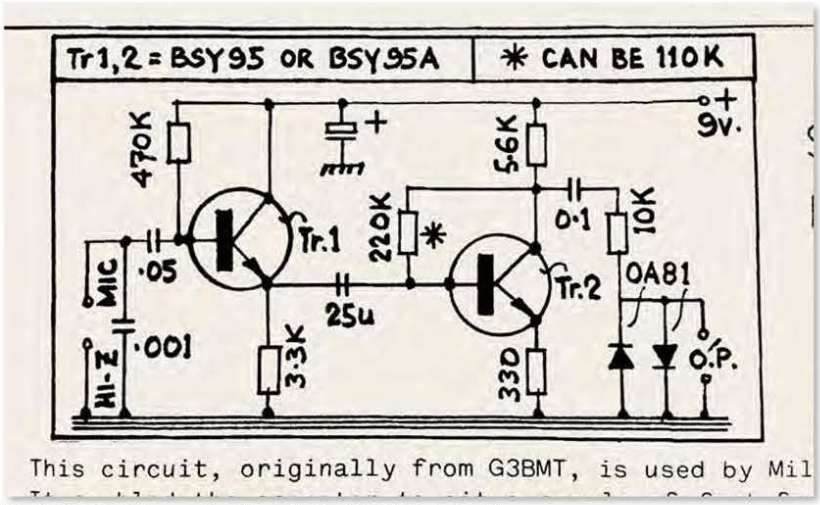
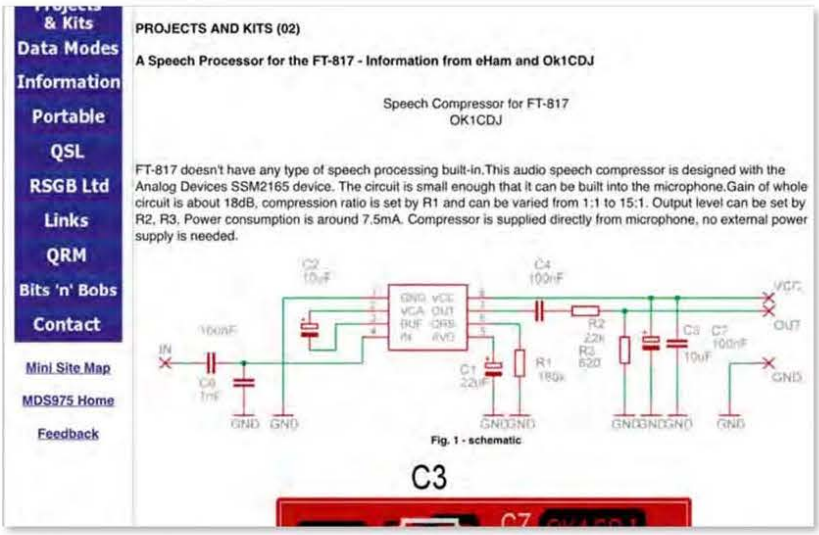


Fig. 2: A 1983-vintage audio clipper design (G-QRP Club circuit).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PROJECTS AND KITS (02)
A Speech Processor for the FT-817 - Information from eHam and Ok1CDJ
Speech Compressor for FT-817
OK1CDJ
FT-817 doesn't have any type of speech processing built-in.This audio speech compressor is designed with the Analog Devices SSM2165 device. The circuit is small enough that it can be built into the microphone.Gain of whole circuit is about 18dB, compression ratio is set by R1 and can be varied from 1:1 to 15:1. Output level can be set by R2, R3. Power consumption is around 7.5mA. Compressor is supplied directly from microphone, no external power supply is needed.
Fig. 1 - schematic
C3

Fig. 3: A compressor based on the SSM2165, designed for use with the FT-817.

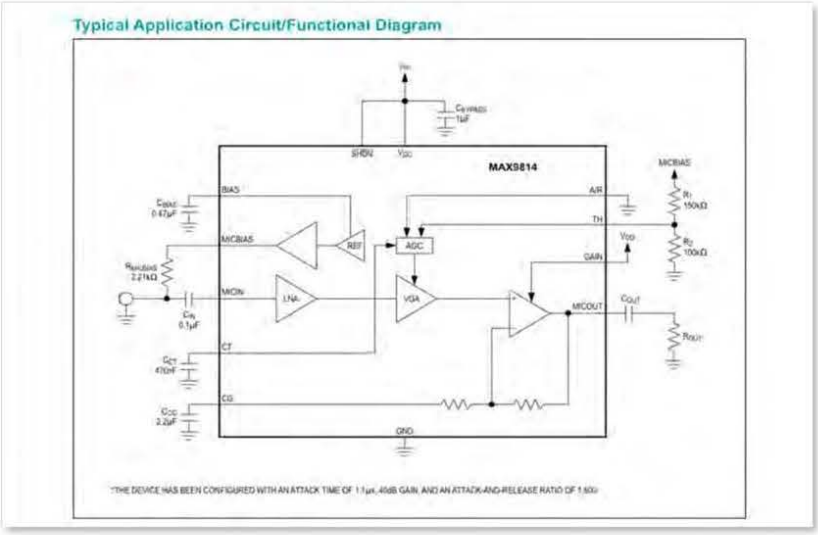

Fig. 4: Featured application from MAX9814 datasheet.

the best design solution (in a pre-digital age). **Fig. 5** shows a Datong ASP SSB speech processor, a product from the 1990s. This creates an SSB waveform at 60kHz (eHam reviews say 'around 450kHz' but that is wrong) from a filtered but otherwise unaltered audio waveform, and clipping is applied to that. The audio is then recovered with another mixer stage and is available at the output to go into any external transmitter at the microphone socket. (There is PTT switching too.) The unwanted harmonics are at 180kHz (minimum) so a 2.7kHz filter can easily get rid of them. 6 to 12dB of clipping (out of 30dB available!) is normally reckoned to offer the best compromise of increased 'oomph' and acceptable sound quality.

(These Datong speech processors are still sought after, by the way. £100 will buy you a nice one but check the battery clips, which often crack and then break off at the connectors.)

In an RF compressor or clipper then, how 'RF' does it need to be? Mostly designers stay in recognisable RF territory but the operating frequency just needs to be high compared to the audio bandwidth. Which gave me an idea…

137kHz is an amateur band and an RF speech processor or clipper design has an oscillator and modulation and demodulation stages. As, with the addition of some RF and AF amplification, does a transceiver. (And switching, lots of filtering and much more, I know – but let's keep this simple for now.) So, a standalone speech processor running at 137kHz, could, in theory, be built with the configurability to be a transceiver in its own right. In theory – more thought is needed I'm sure.

It's not easy to find a design for an RF speech processor nowadays but I found one from YU1OXB on the net. See **Fig. 6** and Useful Links (below). He uses a frequency of 500kHz but, as he says on his website, *"You can use any other frequency crystal filters up to 10 to 11MHz"*. Constructors will see an old friend or two here – note the use of an NE612A, used as a demodulator. This is another obsolete chip but is still used by professionals and amateurs alike (and is, now I think on it, a good subject for another article.)

Fig. 5: The popular Datong speech processor from the 1990s.

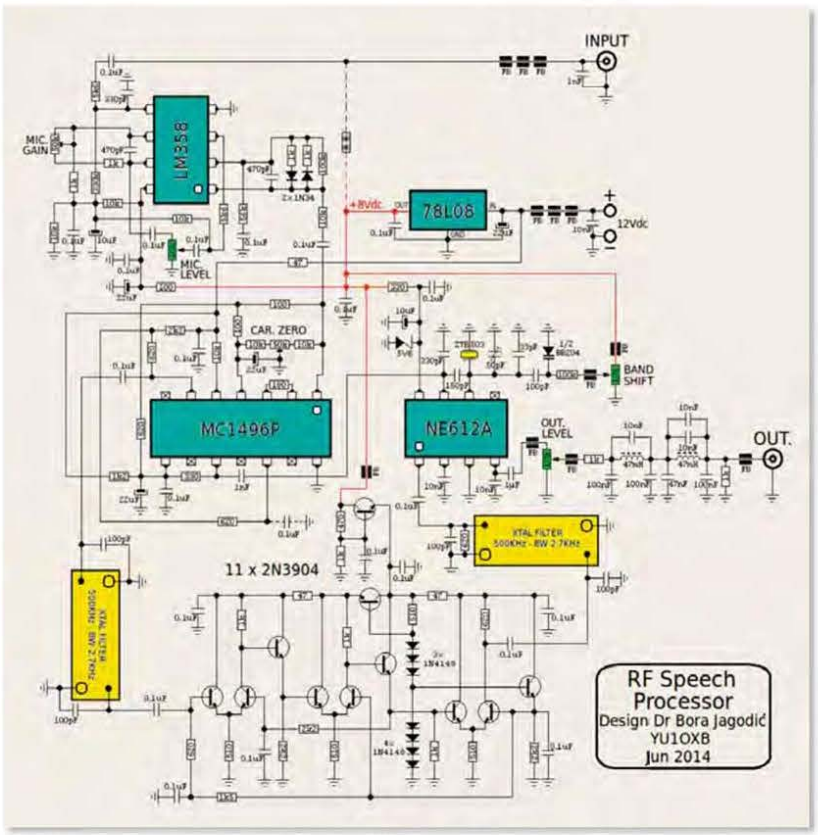

Fig. 6: A recent speech processor design from YU1OXB.

Useful Links

G-QRP Club:
www.gqrp.com
MAX9814 datasheet:
https://datasheets.maximintegrated.com/e/ds/MAX9814.pdf
SSM2167 (replacement for the SSM2155 and 2166) data sheet:
www.mouser.com/ds/2/609/SSM-879586.pdf
Datong ASP SSB instructions:
http://m0ezp.squirrelhouse.biz/images/DatongASP_Ops.pdf
Vocalmaster Speech Processor:
http://kg4jjh.com/vocalmaster.html
YU1OXB's speech processor:
https://forums.qrz.com/index.php?threads/my-version-of-rf-speech-processor.443577